

JT SIP Trunk

Technical Specification

06/10/2026 PUBLIC RELEASE V2.2

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This document describes the technical specification of JT's SIP Trunk solution, utilised for delivering business line voice services in Jersey & Guernsey. This document does not describe practical examples of SIP Trunk nor the implementation a particular system should implement. For further information, the reader may consult the User Guide, or the specifications listed. It is expected that the reader is familiar with basic networking technologies and is aware of the RFC references.

A table of references is included below to list details that are referenced by square brackets and the section of reference, example: [4a].

JT follows SIP Connect v1.1 closely which can be a useful guide for the SIP Trunk service, any references to SIP Connect are indicated by [SC <section>].

1. Stack

Physical [OSI 1] & Data Link Layer [OSI 2]

JT delivers the SIP Trunk at the customer premise as a Cat5 RJ45 Ethernet connection or as a dedicated NNI bearer upon request for cloud-based solutions [1a]. Access to the service will be over JT's private MPLS network or NNI. No internet connection to this service is provided.

Network [OSI 3] & Transport Layer [OSI 4]

Each SIP Trunk will be allocated a single private IPv4 address within a /29 subnet for connectivity from the customers equipment to JT's SBCs exclusively. JT SBCs have public addresses to reduce the risk of addressing conflicts, however, they are not connected to the internet. It is NOT RECOMMENDED to mix both internet and SIP connectivity for SIP Trunk. TCP or UDP are supported for transport.

Application [OSI 7], Signalling & Media

SIP/SDP is used for signalling and RTP/RTCP used for media & reporting respectively. No other application layer is supported. JT supports a selection of standard codecs [2a]. JT provide no support or guarantee of other unlisted codecs working.

2. Authentication & Security

Static Mode is the only trunk authentication mode supported. Authentication is implemented by IP source address (of the PBX). TLS or certificates are not supported. Registration SIP Trunk is no longer supported.

3. Numbering

JT provides multiple DDIs as single numbers and contiguous digit ranges of 10, 100, 1,000 and 10,000.

A pilot number is allocated per trunk for logical identification and billing purposes. It can also be used for calls but is typically not used.

For all calls, the calling and called party MUST be in Global E.164 format. JT will only accept calls in this format.

No explicit support is provided for 00 where + is unable to be dialled or inserted by the customer.

Support is provided for Indirect Dial Access using a prefix (such as 13545). The prefix MUST NOT start with a '+'. It is expected the customer will normalise the dial string ahead of appending the prefix. However, due to known manufacture limitations, this may not be possible. Therefore, JT provides support for non-global E.164 formats.

Called party short codes are allowed such as 123 or 118118, specifically, emergency numbers (999 and 112) MUST be in 3-digit format. No other format is approved for emergencies.

4. Screening

The calling party must be within a number range allocated to the customer by JT for that trunk or trunk group. Calls from numbers outside this range will be blocked as a fraud prevention measure.

Should the customer wish to make an anonymous call, they should do so with the mechanisms specified in [SC 10.2.5]. Further information is included with NICC ND 1439, table 6.1a: mechanisms 1, 2 & 3, table 6.1b: mechanism 1.

The customer SHALL ensure that the A party is not “leaked” to C in From or P-Asserted-Identity for a call forward to an external number, otherwise this will be blocked. JT will remove SIP headers related to call forwards for fraud purposes (such as History-Info).

JT will always present the main number (directory number) to the emergency services¹.

5. Supplementary Services

JT supports a basic set of additional services such as CLI Presentation, CLI Restriction, Call Forwarding Unconditional and Call Forwarding Not Reachable. These are activated & changed by contacting JT. No self-service or short code is provided for these.

6. Trunk Hunting

Trunk Hunting provides transport of calls over multiple physical & logical SIP Trunks to enable geographic & transport redundancy and increased capacity. This can be configured as active/standby or percentage-based load balancing upon request. The customer shall determine how they will route out over multiple trunks. No “hot” redundancy is employed; therefore, a trunk failure will result in the ongoing active calls ending but new sessions can be established over the remaining trunk. Re-route based on error code to INVITE is supported for “503 Service Unavailable” only.

¹ The core network shall derive this from the customers IP address initiating the traffic. The emergency services will use this number for location & call back purposes.

7. SBC Interworking & Resiliency

The customer may access JT's network through three geo-graphically redundant SBCs which have high availability at each site. JT will always aim to have all three SBC's online and processing traffic at any one point. However, this may not be the case in the event of a temporary fault scenario or a long-term site upgrade². Therefore, the customer **MUST** be able to accept ingress traffic from all three sites and is expected to egress traffic to any of the three sites depending on the site availability to use the service effectively and reliably³. The customer **MUST** always ensure that any signalling associated with a call/dialogue, remains on the same SBC and Trunk it was initiated on.

The JT SBCs will determine the health of the trunk based on the response (or lack of) to a SIP OPTIONS message request sent to the customers SIP Trunk allocated IP address. Therefore, the device responsible for the allocated IP address **SHOULD** respond accordingly⁴, advertising its availability to the core if it wishes to participate in sending and receiving calls. The customer **SHALL** determine the health and availability of JT SBCs based on the same mechanism in reverse. No other heartbeat mechanism is supported.

JT do not currently provide a DNS server for SBC IP resolution. The customer may choose to implement their own resolution mechanisms, using DNS, but this is out of scope of this document.

² Due to new security measures, site upgrades may take many weeks, even months in some cases so a prolonged outage should be expected.

³ If the device used to access the SBC only supports primary/secondary there is a risk that traffic will be rejected. To prevent this, JT will advise on site preference & configuration. Please be aware that by only using 2 sites, in a site failure, there will only be 1 site available.

⁴ RFC 3261 section 11.1

Table of reference

OSI Stack and specifications [1a]:

- **[Physical]** RJ-45 CAT5
(TIA/EIA-568)
- **[Data Link]** IEEE 802.3ab
- **[Network]** IPv4 (RFC 791)
- **[Transport]** UDP (RFC 768)
TCP (RFC 793)
MTU 1500
- **[Application]** SIP (RFC 3261)
SDP (RFC 4566)
RTP (RFC 3550)
RTCP (RFC 3550)

Media Codecs [2a]:

- G.711 a-law
- G.711 μ -law
- G.722
- AMR
- AMR-WB
- EVS
- G.711 pass through (fax)
- T.38 (fax)

Called Party formatting for SIP Trunk calling party [3a]:

Global E.164 (examples):

- +447797712345
- +33492944200
- +441534882882

IDA Routing:

- **13545**01534882882
- **13545**+33492944200

Emergency

- **999**
- **112**

Short Codes

- **123**

IP:

A static /29 allocated per trunk with a single assigned to your device per trunk.

SBC signalling address [5a]:

199.34.117.164 (Rue De Pres DC)
199.34.118.164 (Five Oaks DC)
199.34.119.164 (First Tower Lane DC)

SBC address range:

The address below must be whitelisted / allowed. These ranges will contain media services and signalling services.

199.34.117.160/28

199.34.117.176/28

199.34.118.160/28

199.34.118.176/28

199.34.119.160/28

199.34.119.176/28

Ports:

SIP: 5060

Media: Determined by SDP